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1. A sound processing device having means for estimating the amplitude envelope of a sound signal in a plurality of spaced frequency channels, means for analyzing the estimated amplitude envelopes over time so as to detect short-duration amplitude transitions in said envelopes, means for increasing the relative amplitude of said short-duration amplitude transitions, including means for determining a rate of change profile over a predetermined time period of said short-duration amplitude transitions, and means for determining from said rate of change profile the size of an increase in relative amplitude applied to said transitions in said sound signal to assist in perception of low-intensity short-duration speech features in said signal.

2. The device of claim 1, wherein the predetermined time provided is approximately 60ms, and the faster/greater the rate of change, on a logarithmic amplitude scale, of said short-duration amplitude transitions, the greater the increase in relative amplitude which is applied to said transitions.

3. The device of [any preceding] claim 1, wherein the detection of short-duration transitions in the rate of change profile causes up to about 14dB of gain to be applied to the sound signal, the amount of gain being selected according to the nature of the short-duration transition:

(i) a rapid increase followed by a decrease in the profile (a short-duration burst transition) cause a gain increase of up to about 14dB;

(ii) a rapid increase followed by a relatively constant level in the profile (an onset transition) causes a gain increase of up to about 6dB, and

(iii) a relatively constant level followed by a rapid decrease in the profile cause little or no increase in gain.

4. The device of [any preceding] claim 1, including a microphone, a pre-amplifier, a bank of N parallel filters, and means for applying a transient emphasis algorithm to the output of the filters.

5. The device of [any preceding] claim 1, wherein the signal in any filter channel denoted $S_n(t)$, where n denotes the filter channel number and t denotes time, is scaled according to: $S'_n(t) = S_n(t) \times (1 + K_n \times G_n)$, where G_n is the gain factor each channel and K_n is a gain modifier constant equal to about 2.

6. The device of claim 5, wherein a buffer maintains a history of the envelope signal $S_n(t)$ in each filter channel for which an estimate of the slow-varying envelope signal is derived by an averaging window which provides an appropriate equivalence to a 2[nd]-order low-pass filter with a cut off frequency of about 45Hz causing smoothing of the fine profile structure.

7. The device of **[claims]** <u>claim</u> 5 **[or 6]**, wherein the gain factor G_n is related to a function of the 2[nd] order derivative of the slow-varying envelope signal in each filter channel.

8. The device according to **[claims]** <u>claim</u> 5 **[, 6 or 7]**, wherein the gain factor for each filter channel is derived from the function:

$$G_n = (2 \times E_c - 2 \times E_p - E_f) / (E_c + E_p + E_f)$$

where E_c E_p and E_f are estimates of the current, past and future slow-varying envelope signal in each filter channel.

9. The device of claim 8, wherein the additional factor gain G_n applied to the sound signal can range from about 0 to 2 for a slow-varying envelope profile having a rapid rise followed by a rapid fall, about 0 to 0.5 for a profile having a rapid rise followed by a relatively constant level, less than 0.1 for a steady state level followed by a rapid decrease in the profile, and about 0 for a steady state level or slowly varying profile.

10. The device according to **[claims]** <u>claim</u> 8 **[and 9]**, wherein slow-varying envelope profiles exhibiting short-duration peaks of different peak levels but similar peak to valley ratios are amplified by similar amounts.